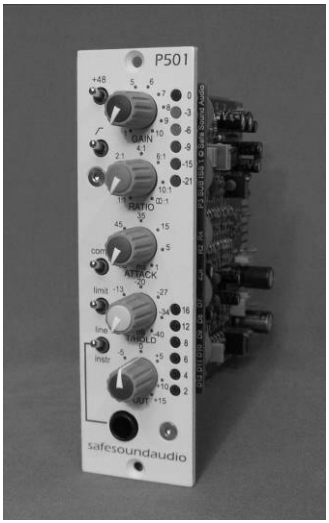

P501 Audio Processor
User Manual

Safe Sound Audio, UK

P501 Audio Processor Features

500 Series
Fully compatible with the API™ 500 Series modular format which allows a number of audio modules to be plugged into rack frames such as the API™ 6-slot Lunchbox and 10-slot 500VPR rack frame.

Input Stage
Wideband 60kHz high gain mic amp is partnered with a front jack accessable balanced line input, which is also relay switchable to become a high impedance DI input. A switchable 80Hz, 18dB/octave hi-pass filter is included.

Compressor
Advanced audio compressor with ***peakride*** which uses multiple side-chains in tandem allowing for very musical dynamic control of source material. *Peakride* compression is very easy to dial in when tracking and evens out levels in a very musical way without robbing the sound of life, so bringing the performance a new energy whilst remaining transparent and not sounding overly affected. It's especially powerful in providing tightly controlled vocal compression with perfect transparency. The inclusion of auto gain make-up allows you to increase gain reduction 'on the fly' without having to constantly readjust the output level.

Variable attack limiter design
Switchable fast limiter offers ultimate protection during tracking or push it hard as a great dynamic effect especially on bass and drums. The limiter features dynamic adjustment of attack time, so the faster the audio front edge, the faster the limiter reacts. Fast clipping protection when tracking but also very useful as a creative tool in its own right when pushed into heavy levels of limiting.

Output stage
The balanced output will drive +22dBu into 600 ohms with a useful ±15dB of output gain control available.

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500 Series Compatibility

The P501 Audio Processor is intended for use within an API™ 500 Series compatible rack frame from which is takes its power and audio connections. It cannot function as a stand-a-lone unit. Information about the API™ 500 Series rack frames can be found at www.apiaudio.com.

As the microphone input stage in the P501 has the capability to switch phantom power, please ensure you choose a rack frame which includes +48V phantom power as this is not generated within the P501.

API is a registered trademark of Automated Processes Inc.

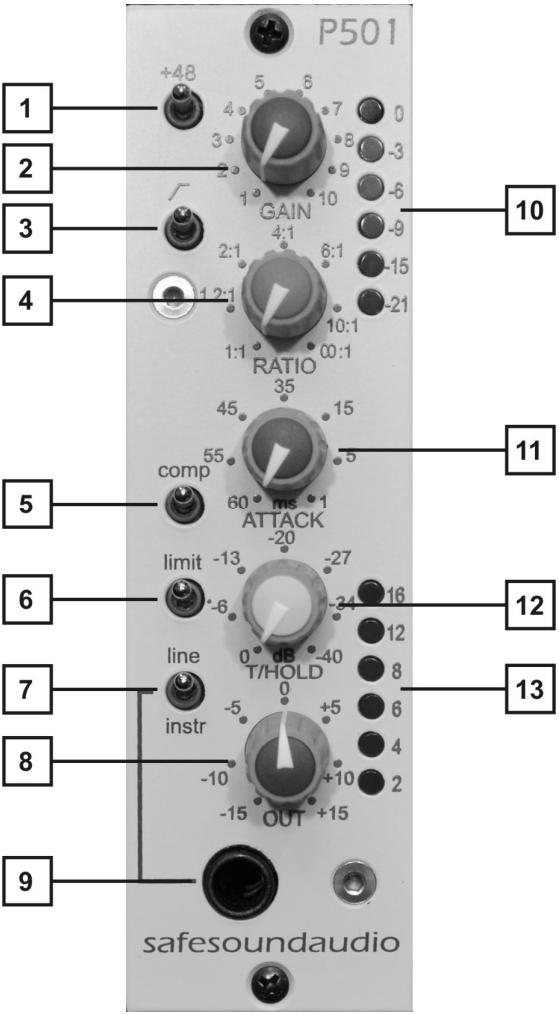
Installation

The P501 is supplied in a foam lined shipping carton and should arrive with you in perfect condition but it is always worth checking the unit for any signs of physical damage before use. Should you have a concern that the unit appears damaged then contact your dealer.

Ensure that your 500 Series compatible rack frame is powered off before inserting the P501 module. Select a spare slot and remove the module mounting screws at that slot location. Slide the P501 module into the slot ensuring that the gold plated edge connector on the P501 aligns with the corresponding connector in the rack frame. Now gently push home the module until the front panel is flush with the rack frame. Use the two screws supplied with the rack frame to secure the module. Never use excessive force when inserting a module into the frame, so if you are struggling to get the module to insert, recheck the alignment and try again. Avoid touching the gold plated edge connector as you could tarnish the connects and reduce the reliability of the audio connection between the module and the rack frame.

Turn on the rack frame power, connect audio and check through the functionality of the P501 module according to the guidance in this manual.

Detailed operational guide



Input Stage

Three audio input types are provided for, these are:

Balanced microphone input, for use with either dynamic or phantom powered condenser microphones. Connect via the XLR input connector for the corresponding channel on the rear of the rack frame. Select switch number **(7)** to '**line**'.

Balanced line level input, for use with any line level source, most commonly when using the P501 as an insert effect within the recording or mixdown chain. Connect via the ¼" jack **(9)** on the front of the P501 module. Switch number **(7)** selected to '**line**'.

Unbalanced instrument input, for direct injection of most electric or bass guitars. This is sometimes called the DI-INPUT (direct injection). Connect via the ¼" jack **(9)** on the front of the P501 module. Switch number **(7)** selected to '**instr**'.

All three input types have different gain ranges available within the P501 to suit the type of source, all controlled from a single gain control **(2)**.

+48V Phantom power is selected on by selecting switch **(1)** downwards. Always connect the microphone before switching on phantom power and always switch off phantom power before disconnecting the microphone. Always best to mute your monitoring chain first as switching phantom power cause audible 'thumps' through your audio chain.

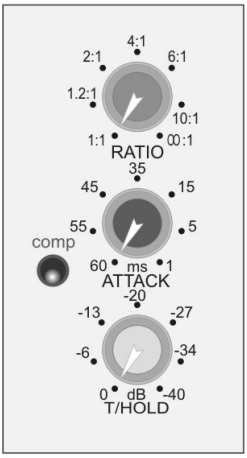
High Pass Filter

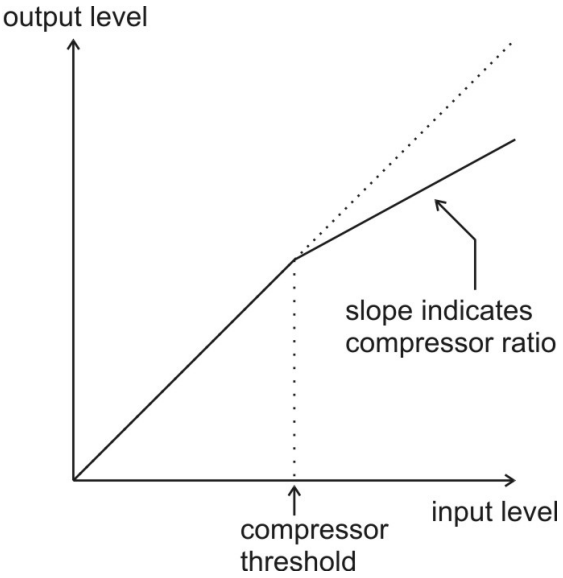
The high pass filter is sited in the audio signal path immediately after the input stage but before the compressor and limiter. It operates at 80Hz with a slope of 18dB/octave.

Bring the high pass filter into circuit by selecting switch **(3)** downwards.

Peakride **Compressor**

Compressors progressively reduce the gain of an audio signal as its input level rises above the compressor threshold as shown in the diagram below.





The *compressor threshold* is the level point in dB above which the audio gain will be reduced. On the P501 this can be varied from 0dB to -40dB using the T/HOLD control **(12)**.

The *ratio control* **(4)** varies the degree of gain reduction which is applied, from 1:1 (no effect) to ∞ : 1 which will make the compressor act like a limiter.

The *attack time* is how quickly the compressor will react to audio which rises above the threshold point. In the P501 the attack time can be varied from 60ms (slow) to 1ms (fast) using the ATTACK control **(11)**. The setting of the attack time usually has a large impact on the compression effect.

The *release time* is how quickly the compression effect is removed when the audio falls back below the threshold. The P501's release time is programmed to follow the dynamics of the falling audio level. This is an important aspect of the P501's compressor design and offers a very natural sounding form of compression.

The amount of gain reduction (due to the compressor and limiter) is shown in the 6 segment LED meter **(13)**.

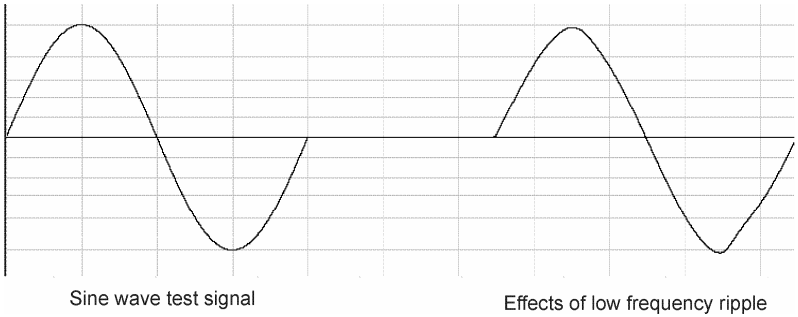
Bring the compressor into circuit by selecting switch **(5)** downwards.

Because compression is a gain reduction tool, it will tend to lower the maximum audio level through the audio chain. Make-up gain is used to replace this 'lost' level and in the case of the P501 this is done automatically; the amount of make-up gain applied being related to the threshold and ratio setting. Auto gain make-up is really useful as it allows you to increase gain reduction 'on the fly' without having to constantly readjust the output level.

Peakride **Compressor design theory**

With so many analogue and digital compressors on the market today, we decided to try a new approach to single audio band compression which gives most of the advantages of multi-band compressors (plus a few more!) without the operational complexity of band splitting. The full story of *peakride* compression is told in our design *white paper* but here are the highlights;

It's desirable for a compressor to be able to offer a wide range of attack and release times to suit a variety of source material. For example, percussion typically requires medium fast attack and release times whereas some vocals require quite fast attack times and medium/slow release times. Fast attack, fast release compressors often have poor audio performance especially distortion caused by ripple of the side-chain by the audio. Have a look at the diagram below;



Notice that the pure tone audio signal is beginning to lose its smooth top and bottom curves and begins to resemble a triangular waveform. This is not just an issue for bass guitar and bass drum compression. Many audio sources including piano and vocals have either a primary low frequency 'carrier' component and vibrato/tremolo induced low frequency components. All can cause audible distortion when being compressed or limited.

The *peakride* design overcomes these problems by the use of three separate control side-chains which act in the time domain (rather than in the frequency domain of a multi-band compressor).

Above the compressor's threshold;

Short duration audio bursts (such as percussion) get processed with the fast attack fast release compression which they tend to require.

Longer duration audio bursts get processed with an initial attack time as set on the front panel and an initial short release time but; as the audio sample continues, the attack and release times are progressively lengthened according to two factors;

- The time the audio sample has been above the compressor threshold
- The continuing dynamics of the audio signal

A second problem with traditional fast attack compressor designs is caused by a misunderstanding between the desirability for fast attack, especially to provide adequate control of certain vocal styles,

and the usual consequence that the whole vocal phrase then suffers from over compression. It is often difficult to achieve the attack speed without the undesirable over compression.

The *peakride* compressor achieves this by altering the ratio of the compressor dynamically. So you can set an average compression ratio on the front panel control but the actual delivered ratio will alter to suit the dynamics of the audio as it rises above the threshold level.

These principles are at the heart of the peakride design and are achieved by mixing the three compressor side-chains at different levels and with different compression ratios. In effect, the compressor side-chain is able to ride the peaks of the audio, reacting quickly but smoothly to fast attack peaks, and then able to track the dynamics of the continuing audio till it falls below the threshold point.

Limiter

The P501 includes a switchable audio limiter which can be used to prevent digital clipping in your recording chain. The limiter is sited in the audio signal path immediately before the output level control and operates at fixed threshold of +6dBu (nominal).

The limiter features dynamic adjustment of attack time, so the faster the audio front edge, the faster the limiter reacts. So fast clipping protection when tracking but also very useful as a creative tool in its own right when pushed into heavy levels of limiting.

We've added a programme tracking release circuit which provides for very smooth recovery from limiting, adapting to the natural dynamics of the audio source.

Bring the limiter into circuit by selecting switch **(6)** downwards.

Output Gain and Level Metering

The output gain control **(8)** is sited in the signal path immediately after the limiter and just before the final balanced output stage which in turn feeds the audio output to the selected channel output XLR in the rack frame. The most common use of the output gain control is to match the peak **output** level of the P501 to the peak **input** level of your recording device. Once you have done this level matching it's usually not necessary to alter the output gain control during recording.

The P501's level meter **(10)** is intentionally placed BEFORE the output gain control. We suggest you use the level meter in conjunction with the INPUT gain control during recording set-up so that the maximum audio level during tracking is lighting one or both of the yellow level LED's but not intentionally lighting the red level LED. When switched into circuit the P501's limiter will keep the maximum output level below 0dB (nominally +6dBu).

Operating at higher nominal recording levels (e.g. +18dBu) is also possible by switching the P501 limiter off and using only the recording meters in your recording device to monitor the maximum desired recording level.

Technical Specification

P501 AUDIO PROCESSOR

Physical

Size :	API™ 500 Series Compatible (38mm wide by 133mm high by 150mm deep)
Weight :	0.4kg (0.9lbs)
Power requirements :	±16 volts dc at ±120mA to the standard API 500 Series pin-outs. +48 volts dc for phantom power

Fusing :	On board ‘auto reset’ polyfuses protecting the ±16V and +48V rails

Main processor audio path

Frequency response :	-0.5dB points at 5Hz and 60kHz
Distortion :	< 0.01% at 1kHz (typically 0.008%) main signal path working at 0dBu

Maximum input levels

Microphone :	+12dBu
Line input :	+22dBu
Instrument :	+18dBu (when fed unbalanced)

Input impedances

Microphone :	2k4 ohms
Line input :	> 10k ohms
Instrument :	1M ohms

Equivalent input noise figures

Microphone :	128dB (max gain with 150 ohms source)
Line input :	100dB (unity gain)
Instrument:	105dB (max gain)

High Pass Filter

Switchable 80Hz hi-pass filter with 18dB/octave slope

Balanced Line Output

Output Gain :	±15dB (post limiter)
Maximum Output Level (post limiter) :	+22dBu
Output impedance:	50 ohms

Output Noise :	-93dBu RMS unweighted, measured 22Hz to 22kHz
	-95dBu RMS A-weighted, measured 22Hz to 22kHz

Input Gain Ranges

Microphone input :	0dB to +72dB
Line input :	-10dB to +20dB
Instrument input :	+2dB to +33dB

***Peakride* Compressor**

Threshold range :	0dB to -40dB
Attack time range :	60ms to 1ms
Release time :	variable from 90ms to 500ms (tracks the natural audio decay)
Ratio :	1:1 to INF:1
Make-up gain :	automatically set in proportion to the threshold and ratio settings
Gain reduction meter :	calibrated in dB's

Dynamic Attack Limiter

Threshold :	fixed internally to +6dBu (nominal)
Attack time :	audio related with a fastest response of 0.5ms
Release time :	audio related from 50ms to 500ms

Metering

Level metering :	6 element LED display measuring -21dB to 0dB (0dB equates to Limiter Threshold)
Gain Reduction :	6 element LED display

Warranty

Safe Sound Audio provides a 12 month warranty from the purchase date according to the following conditions;

This warranty is not transferable and applies only to the original purchaser of the unit.

If the unit should become faulty, then contact Safe Sound Audio or your local distributor for a Returns Authorisation Number. No items will be accepted for warranty repair without this authorisation number.

You must be able to produce proof of the purchase date.

If the returned unit should prove faulty then, at Safe Sound Audio's choice, we will either repair or replace the unit.

The customer is responsible for the cost of sending the unit back to Safe Sound Audio or our authorised distributor including insurance of the unit during shipping.

Safe Sound Audio or our authorised distributor will be responsible for the cost of shipping the repaired or replaced unit back to the customer including insurance of the unit during shipping.

The warranty will be void if the unit has;

- Suffered physical damage.
- Been repaired or modified by anyone other than Safe Sound Audio or its authorised representative.
- Has been connected to an incorrect source of power.
- Has been damaged due to liquid spillage.

Safe Sound Audio shall not be liable for any special or consequential damages resulting from the use of this product.

How to contact us

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Safe Sound Audio reserves the right to make changes and improvements to the design of this product without notice.

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